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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/783,572	Applicant(s) KUBLER ET AL.	
	Examiner BO HUI A. ZHU	Art Unit 2465	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *18 February 2010*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *22-69* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *22-69* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Continued Examination Under 37 CFR 1.114

1. A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 02/18/2010 has been entered.

Response to Amendment

2. The amendment filed on 02/18/2010 has been entered.

Claims 22 - 69 are pending.

Claims 22 – 69 are rejected.

Claim Rejections - 35 USC § 103

3. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4. Claims 22, 25-28, 31, 32 and 34 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kennedy, III et al (U.S Patent No. 5,734,981) in view of Sherif (U.S. Patent No. 5,459,722).

Regarding claim 22, Kennedy, III et al teach a device (18, FIG. 1) for communicatively coupling a packet network (16, FIG. 1) to at least one communication network (38, FIG. 1) having a different information format, the device comprising: a packet interface (160, FIG. 3) for exchanging information via file packet network (column 4, lines 56-61); at least one network interface (170, 172, FIG. 3), each of the at least one network interface for exchanging information via an associated one of the at least one communication network in an associated format (column 11, lines 52-56); a controller (166, FIG. 3) for receiving call setup information (call delivery information) from one of the packet network (16, FIG. 1) and the at least one network interface (column 11, lines 33-35), the controller adapting the operation of a converter and establishing an association between the packet interface (160, FIG. 3) and one of the at least one network interface (170, FIG. 3), based upon the call setup information (column 11, lines 48-51, column 12, lines 50-54).

Kennedy III et al. does not disclose at least one converter for selectively converting information received by the packet interface for transmission via one of the at least one network interface in the associated format, and for selectively converting for transmission via the packet interface information received from the one of the at least one network interface in the associated format; and a controller for receiving from either of the packet network and the at least one network interface, signaling information that

initiates a call connection between the packet network and one of the at least one communication network, the controller adapting the operation of a converter and establishing the call connection between the packet network and the one of the at least one communication network, based upon the received signaling information and a cross-reference between an address on the at least one communication network and an associated address on the packet network.

Sherif teaches a converter (Fig. 1, 30 and 40) for selectively converting information received by a packet interface (Fig. 1, 70) for transmission via one network interface (Fig. 1, 10) in the associated format, and for selectively converting for transmission via the packet interface (Fig. 1, 70) information received from the one network interface (Fig. 1, 10) in the associated format; a controller (Fig. 6, 230) for receiving from either of the packet network and the at least one network interface, signaling information that initiates a call connection between the packet network and one of the at least one communication network, the controller adapting the operation of a converter and establishing the call connection between the packet network and the one of the at least one communication network, based upon the received signaling information and a cross-reference between an address on the at least one communication network and an associated address on the packet network (column 5, line 41 – 46)

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the system of Kennedy III et al. to include the elements of at least one converter for selectively converting information received by the packet

interface for transmission via one of the at least one network interface in the associated format, and for selectively converting for transmission via the packet interface information received from the one of the at least one network interface in the associated format; and a controller for receiving from either of the packet network and the at least one network interface, signaling information that initiates a call connection between the packet network and one of the at least one communication network, the controller adapting the operation of a converter and establishing the call connection between the packet network and the one of the at least one communication network, based upon the received signaling information and a cross-reference between an address on the at least one communication network and an associated address on the packet network as shown in Sherif in order to allow for connectivity between the PSTN network and ATM networks.

Regarding claim 25, Kennedy, III et al teach the device of claim 22 wherein the information exchanged via the packet interface comprises digitized voice information (column 9, lines 65-67, column 10, lines 1-4).

Regarding claim 26, Kennedy, III et al teach the device of claim 25 wherein at least a portion of the information exchanged via the packet interface is unrelated to the exchange of digitized voice information (column 10, lines 46-55).

Regarding claim 27, Kennedy, III et al teach the device of claim 22 wherein the at least one network interface (170, FIG. 3) provides the functionality of a conventional telephone switching network interface (column 11, line 48).

Regarding claim 28, Kennedy, III et al teach the device of claim 27 wherein the at least one network interface provides at least one of a battery supply, over-voltage protection, ringing current, tone generation, tone detection, two wire to four wire conversion, and test functionality (262, 264, 265, FIG. 6, column 13, lines 6-12).

Regarding claim 31, Kennedy, III et al teach the device of claim 27 wherein the at least one network interface is a digital interface (20, FIG. 1, column 7, lines 34-42, column 8, lines 6-9).

Regarding claim 32, the device of claim 22 wherein the at least one network interface is a second packet interface (172, FIG. 3, column 11, lines 51-52).

Regarding claim 34, Kennedy, III et al teach the device of claim 22 wherein the at least one converter adapts information received via the packet interface into analog modem signals (174, FIG. 3) for transmission via the at least one network interface (column 12, lines 39-45, 50-52), and adapts analog modem signals received via the at least one network interface into information for transmission via the packet interface (column 12, lines 55-61).

5. Claims 23, 24, 29 and 33 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kennedy, III et al (U.S Patent No. 5,734,981) in view of Sherif (U.S. Patent No. 5,459,722) and further in view of Henley et al (U.S Patent No. 5,526,353).

Regarding claims 23 and 24, Kennedy III et al. in view of Sherif teaches the device of claim 22, respectively.

However, Kennedy, III et al fail to explicitly teach the packet interface is compliant with an Internet protocol (IP) and the Internet Protocol (IP) further comprises the transmission control protocol (TCP)/Internet protocol (IP).

Henley et al disclose a system and method for communication of audio data over a packet-based network. The teaching recite Transmission Control Protocol/Internet Protocol (TCP/IP) is one of the supported network and transport protocols (column 4, lines 6-7). Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al make the packet interface compliant with IP and to include TCP/IP as a transport protocol in the call delivery system as taught by Henley et al. One is motivated as such to employ error and flow control in order to realize significant loss of throughput in packet retransmissions (column 4, lines 7-14).

Regarding claim 29, Kennedy, III et al in view of Sherif teach the device of claims 27.

Kennedy, III et al however, fail to teach the at least one converter converts digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information.

Henley et al teach a system and method for communication of audio data over a packet-based network. The system according to the embodiment consist of a decompression/analog conversion circuit for converting a stream of digital audio data to analog audio signal (column 7, lines 27-31) and a digital compression circuit for converting analog audio signal into a stream of digital audio data (column 7, lines 19-

21). Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al to have the at least one converter enabled the conversions of digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information as taught by Henley et al. One is motivated as such to compensate for jitter in a computer network in order to provide high fidelity transmission of audio data through the network (column 4, lines 66-67).

Regarding claim 33, Kennedy, III et al in view of Sherif teach the device of claim 22.

However, Kennedy, III et al fail to teach the at least one converter compensates for a difference in bit rate between interfaces.

Henley et al teach a system and method for communication of audio data over a packet-based network. The system according to the embodiment consists of a decimation circuit adapted to detect when the buffer is too long and adjusts the buffer toward its predetermined length. This happens when the clock of a coder/decoder (CODEC) triggers too slowly or if the audio data are transmitted at an excessive rate through the LAN, thus data are read from the buffer slower than they are written to the buffer (column 5, lines 65-67, column 6, lines 1-8). Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al to have the at least one converter enabled the conversions of digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information as taught by Henley et al. One is motivated

as such to ensure the buffer stays close to it predetermined length for efficient realignment of audio data in the buffer (column 6, lines 11-14).

6. Claim 30 is rejected under 35 U.S.C. 103(a) as being unpatentable over Kennedy, III et al (U.S Patent No. 5,734,981) in view of Sherif (U.S. Patent No. 5,459,722) and Henley et al (U.S Patent No. 5,526,353) further in view of Sharman (U.S Patent No. 5,774,854).

Regarding claim 30, Kennedy, III et al teach the device of claim 29.

Kennedy, III et al however, fail to teach the converting comprises buffering of digitized voice information for a period of time to minimize gaps in an analog voice signal.

Sharman teaches a text to speech system operating in real using an acoustic processor and a linguistic processor. Due to the computational time the linguistic processor requires to process data, future requests from the acoustic processor cannot be made. Thus gaps in the speech output often occur when the acoustic processor requests data from the linguistic processor. Sharman proposes a solution to overcome the gaps in data by adjusting the buffer for minimal of output data so that future requests can be supplied in a timely manner (column 7, lines 39-48). Hence the propagation delay caused by the linguistic processor is a factor affecting the adjustment in the buffer for desired optimal output. Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al to have the converting comprised buffering of digitized voice

information for a period of time to minimize gaps in an analog voice signal as taught by Sharman. One is motivated as such to accurately halt the system based on the output in the event that an interruption occurs (abstract, column 2, lines 34-39).

7. Claims 35, 38 – 41, 44 – 47, 51, 54 – 57 and 60 - 63 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kennedy, III et al (U.S Patent No. 5,734,981) in view of Sherif (U.S. Patent No. 5,459,722).

Regarding claims 35 and 51, Kennedy, III et al teach a system and a method for communicatively coupling a packet network (16, FIG. 1) to at least one communication network (38, FIG. 1) having a different information format, the method comprising: receiving call setup information (call delivery information) from one of the packet network (16, FIG. 1) and the at least one communication network (column 3, lines 40-43); establishing an association between the packet network (16, FIG. 1) and one of the at least one communication network (38, FIG. 1) based upon the call setup information (column 8, lines 4-9).

Kennedy III et al. does not disclose at least one converter for selectively converting information received by the packet interface for transmission via one of the at least one network interface in the associated format, and for selectively converting for transmission via the packet interface information received from the one of the at least one network interface in the associated format; and a controller for receiving from either of the packet network and the at least one network interface, signaling information that initiates a call connection between the packet network and one of the at least one

communication network, the controller adapting the operation of a converter and establishing the call connection between the packet network and the one of the at least one communication network, based upon the received signaling information and a cross-reference between an address on the at least one communication network and an associated address on the packet network.

Sherif teaches a converter (Fig. 1, 30 and 40) for selectively converting information received by a packet interface (Fig. 1, 70) for transmission via one network interface (Fig. 1, 10) in the associated format, and for selectively converting for transmission via the packet interface (Fig. 1, 70) information received from the one network interface (Fig. 1, 10) in the associated format; a controller (Fig. 6, 230) for receiving from either of the packet network and the at least one network interface, signaling information that initiates a call connection between the packet network and one of the at least one communication network, the controller adapting the operation of a converter and establishing the call connection between the packet network and the one of the at least one communication network, based upon the received signaling information and a cross-reference between an address on the at least one communication network and an associated address on the packet network (column 5, line 41 – 46)

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the system of Kennedy III et al. to include the elements of at least one converter for selectively converting information received by the packet interface for transmission via one of the at least one network interface in the associated

format, and for selectively converting for transmission via the packet interface information received from the one of the at least one network interface in the associated format; and a controller for receiving from either of the packet network and the at least one network interface, signaling information that initiates a call connection between the packet network and one of the at least one communication network, the controller adapting the operation of a converter and establishing the call connection between the packet network and the one of the at least one communication network, based upon the received signaling information and a cross-reference between an address on the at least one communication network and an associated address on the packet network as shown in Sherif in order to allow for connectivity between the PSTN network and ATM networks.

Regarding claims 38 and 54, Kennedy, II et al teach the method of claim 35 wherein the information exchanged via the packet network comprises digitized voice information (column 9, lines 65-67, column 10, lines 1-4).

Regarding claims 39 and 55, Kennedy, II et al teach the method of claim 35 wherein the information exchanged via the packet network comprises data (column 10, lines 39- 44).

Regarding claims 40 and 56, Kennedy, II et al teach the method of claim 39 wherein at least a portion of the data is unrelated to the exchange of digitized voice information (column 10, lines 46-55).

Regarding claims 41 and 57, Kennedy, II et al teach the method of claim 35 wherein the at least one communication network is a second packet network (172, FIG. 3, column 11, lines 51-52).

Regarding claims 44 and 60, Kennedy, II et al teach the method of claim 35 wherein the at least one communication network comprises a conventional telephone switching network (38, FIG. 1, column 6, lines 7-10).

Regarding claims 45 and 61, Kennedy, II et al teach the method of claim 44 wherein the second information format is an analog format (column 12, lines 55-56; PSTN 38 can include traditional landline telephone adapted to making analog phone calls - column 6, lines 8-9).

Regarding claims 46 and 62, Kennedy, III et al teach the method of claim 44 wherein one of the second information format is a modem signal (column 12, lines 39-42).

Regarding claims 47 and 63, Kennedy, III et al teach the method of claim 44 wherein the second information format is a digital format (column 6, lines 15-18; caller 40 can make calls from network 41, which can be a personal communication service (PCS) network supporting digital format).

8. Claims 36, 37, 42, 43, 48, 49, 52, 53, 58, 59, 64 and 65 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kennedy, III et al (U.S Patent No. 5,734,981) in view of Sherif (U.S. Patent No. 5,459,722) and further in view of Henley et al (U.S Patent No. 5,526,353).

Regarding claims 36, 37, 42, 43, 52, 53, 58 and 59, Kennedy, III et al teach the device of claims 35 and 51, respectively.

However, Kennedy, III et al fail to explicitly teach the packet interface is compliant with an Internet protocol (IP) and the Internet Protocol (IP) further comprises the transmission control protocol (TCP)/Internet protocol (IP).

Henley et al disclose a system and method for communication of audio data over a packet-based network. The teaching recite Transmission Control Protocol/Internet Protocol (TCP/IP) is one of the supported network and transport protocols (column 4, lines 6-7). Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al make the packet interface compliant with IP and to include TCP/IP as a transport protocol in the call delivery system as taught by Henley et al. One is motivated as such to employ error and flow control in order to realize significant loss of throughput in packet retransmissions (column 4, lines 7-14).

Regarding claims 48 and 64, Kennedy, III et al teach the device of claims 35 and 51,

Kennedy, III et al however, fail to teach the at least one converter converts digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information.

Henley et al teach a system and method for communication of audio data over a packet-based network. The system according to the embodiment consist of a decompression/analog conversion circuit for converting a stream of digital audio data to

analog audio signal (column 7, lines 27-31) and a digital compression circuit for converting analog audio signal into a stream of digital audio data (column 7, lines 19-21). Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al to have the at least one converter enabled the conversions of digitized voice information into an analog voice signal, and an analog voice signal into digitized voice information as taught by Henley et al. One is motivated as such to compensate for jitter in a computer network in order to provide high fidelity transmission of audio data through the network (column 4, lines 66-67).

Regarding claims 49 and 65, Kennedy, III et al teach the method of claims 35 and 41.

Kennedy, III et al fail to explicitly teach the transforming comprises converting an analog voice signal into digitized voice information.

Henley et al teach a system and method for communication of audio data over a packet-based network. The system according to the embodiment consists of a digital compression circuit for converting analog audio signal into a stream of digital audio data (column 7, lines 19-21). Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al to have the transformation of an analog voice signal to digitized voice information as taught by Henley et al. One is motivated as such to compensate for jitter in a computer network in order to provide high fidelity transmission of audio data through the network (column 4, lines 66-67).

9. Claims 50 and 66 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kennedy, III et al (U.S Patent No. 5,734,981) in view of Sherif (U.S. Patent No. 5,459,722) and further in view of Sharman (U.S Patent No. 5,774,854).

Regarding claims 50 and 66, Kennedy, III et al teach the method of claims 35 and 41.

Kennedy, III et al however, fail to teach the converting comprises buffering of digitized voice information for a period of time to minimize gaps in an analog voice signal.

Sharman teaches a text to speech system operating in real using an acoustic processor and a linguistic processor. Due to the computational time the linguistic processor requires to process data, future requests from the acoustic processor cannot be made. Thus gaps in the speech output often occur when the acoustic processor requests data from the linguistic processor. Sharman proposes a solution to overcome the gaps in data by adjusting the buffer for minimal of output data so that future requests can be supplied in a timely manner (column 7, lines 39-48). Hence the propagation delay caused by the linguistic processor is a factor affecting the adjustment in the buffer for desired optimal output. Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al to have the converting comprised buffering of digitized voice information for a period of time to minimize gaps in an analog voice signal as taught by

Sharman. One is motivated as such to accurately halt the system based on the output in the event that an interruption occurs (abstract, column 2, lines 34-39).

10. Claims 67 - 69 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kennedy, III et al (U.S Patent No. 5,734,981) in view of Sherif (U.S. Patent No. 5,459,722) and further in view of Bates et al. (US Patent No. 5,239,577).

Regarding claim s 67 – 69, Kennedy III et al. in view of Sherif discloses all the elements as discussed above.

However, Kennedy III et al. does not disclose the address on the at least one communication network comprises a telephone number.

Bates et al. teaches a communication network comprises a telephone number (e.g. see column 7, line 44 – 64).

It would have been obvious to one of ordinary skill in the art at the time the invention was made to modify the system of Kennedy III et al. to include the element of the address on the at least one communication network comprises a telephone number as shown in Bates et al. in order to allow communication using a telephone number.

Response to Arguments

11. Applicants' arguments in regards to independent claims 22, 35 and 51 have been fully considered but they are moot in view of the new grounds of rejections.

Applicants' arguments in regards to dependent claims 30, 50 and 66 have been fully considered but they are not persuasive. Applicants argue that the cited prior arts do not disclose buffering digitized voice information for a period of time to minimize gaps in an analog voice signal. In particular, applicants argue that the teaching of buffering "annotated text segments" is different from the buffering of "digitized voice information." (Remarks, page 24 – 26, page 29 – 31). Examiner respectfully disagrees. Sharman teaches that due to the computational time the linguistic processor requires to process data, future requests from the acoustic processor cannot be made. Thus gaps in the speech output often occur when the acoustic processor requests data from the linguistic processor. Sharman proposes a solution to overcome the gaps in data by adjusting the buffer for minimal of output data so that future requests can be supplied in a timely manner (column 7, lines 39-48). Hence the propagation delay caused by the linguistic processor is a factor affecting the adjustment in the buffer for desired optimal output. Therefore, it would have been obvious to one with ordinary skill in the art at the time of the invention was made to modify the teaching of Kennedy, III et al to have the converting comprised buffering of digitized voice information for a period of time to minimize gaps in an analog voice signal as taught by Sharman. One is motivated as such to accurately halt the system based on the output in the event that an interruption occurs (abstract, column 2, lines 34-39).

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to BO HUI A. ZHU whose telephone number is (571)-270-1086. The examiner can normally be reached on Mon-Thu 10am-6pm.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Jay Patel can be reached on (571)-272-2988. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/B. A. Z./
Examiner, Art Unit 2465

/Jayanti K. Patel/
Supervisory Patent Examiner, Art Unit 2465